Link Layer Design for a Military Narrowband Radio Network

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ABSTRACT

We are proposing a link layer for a new VHF/UHF narrowband (25 kHz) waveform (NBWF) to be standardised by NATO for combat net radio (CNR). The user requirements are for a networking waveform providing concurrent voice and IP data. The primary voice service is push-to-talk (PTT) multicast to reach all or a predefined group of users in a multi-hop network. The primary vocoder to be used is MELPe, but other vocoders are also required to be supported. We will try to serve multicast voice by reserving resources only when required, including any required automatic relays. Data transfer relies on a combination of contention and reservation. The paper will outline the proposed link layer, with focus on the most challenging services. A simulator is being developed, based on Omnet++. Some results for the performance (delay and probability of success) of multicast voice are presented, in addition to initial performance for data services. Simulation results are compared to analytical estimates.

1.0   INTRODUCTION

Currently, there is an action within NATO Sub-committee 6 Ad hoc working group 2 (SC/6-AHWG/2) to produce a networking waveform standard for VHF/UHF radio communications between coalition land forces, termed NarrowBand WaveForm (NBWF)[1]. The first level of ambition is to reach agreement on a 25 kHz non-EPM[1] waveform that supports concurrent voice and IP data. Later levels of ambition include adding EPM capabilities and larger bandwidth.

There have been many similar attempts to standardize a common digital waveform for Combat Net Radio (CNR), both in EUROCOM and NATO. So far these attempts have stranded on protectionism. The Software Defined Radios of tomorrow may render the current standardisation attempt possible. Future radios will be able to run both proprietary and standardised waveforms and switch between these waveforms on the fly.

The physical layer (modulation and coding) has been the focus of the working group until recently. We (FFI) are currently working on a proposal for the link layer: MAC (Medium Access Control) and LLC (Logical Link Control). It is designed to build upon a physical layer (PHY) proposal by CRC, Canada [5]. The PHY offers data rates from 20 to 96 kbps. Our initial work was reported in [2]. We will first revisit the requirements, discuss design priorities and trade-offs before presenting an outline of our proposed link layer. Finally, we present some initial simulation results.

2.0   REQUIREMENTS AND PREREQUISITES

The requirements to the NBWF [1] were addressed in [2], but we will give an updated summary here. The operational requirements are for a CNR waveform to serve both voice and IP data services concurrently (unicast and multicast) in a coalition network. There is a requirement to address up to 250 nodes, but the

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[1] Electronic Protection Measures are usually obtained by frequency hopping to counter jammers.

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14. ABSTRACT

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typical number of active radios in a network is more likely to be less than 50, maybe even less than 25. The principal services to support will be multicast voice (push to talk - PTT), situation awareness (SA) and radio based combat ID (RBCI) [3]. In addition, the waveform must be able to support other services such as targeting, core services and functional services.

There is a requirement for multi-hop distribution of all services, but typically CNR networks are such that most radios are within a single radio hop at low data rates. This means that spatial slot reuse is not very feasible for our MAC protocol.

The following functional requirements and assumptions are used as a basis for designing the link layer protocol:

- A significant fraction of the traffic is radio-broadcast (single hop) or multicast, both voice and data.

- Traffic consists of a mixture of predictable or streaming type and more random type traffic.

- Voice is prioritized and must be served with short delay and small jitter.

- Voice is primarily coded as MELPe at 2.4 kbps [4], but other vocoders shall be supported.

- The primary PHY rate of 20 kbps[2] is normally used for all signalling and multicast traffic, in order to achieve the best possible single-hop coverage..

- We assume that the network has been established and focus our study on an operational network.

The structure of the physical layer waveform is as follows, with preliminary values indicated (see figure 1):

- a synchronisation preamble ($T_{pre} \approx 1.5$ ms) and

- a Start_Of_Message ($T_{SOM} \approx 2.1$ ms), followed by

- a 12 bit parameter field Par ($T_{par} \approx 1.6$ ms) with PHY protocol control information (PCI) (strongly coded),

- an optional delay (T_{TR}) is required to change modulation rate,

- an optional midamble (Mid) for channel equalization, and finally

- the higher layer PCI and payload in one or more interleaver blocks, optionally separated by new midambles.

As the preamble and SOM are used by MAC to detect ongoing transmissions it must be strong enough (low signal to noise ratio – SNR) to be detected properly at any data rate. The same strength requirement applies to the Par.
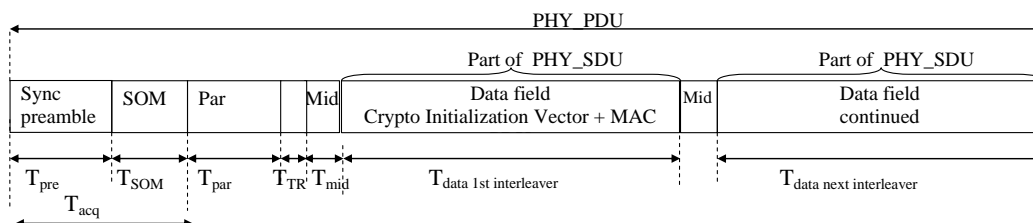

Figure 1: The structure of the physical layer waveform

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[2] Using a symbol rate of 30 ksymbols/s and a 2/3 coder.

Information is transferred in interleaver blocks using serially concatenated continuous-phase coded modulation (CPM) [5], [6]. For this modulation, the SNR performance depends on the length of the interleaver block. The shortest useful block length is probably around 5 ms at the lowest rate without sacrificing too much noise resilience.

The efficiency of a multiple access radio system depends on the time needed for signalling relative to the information transfer time. With strong delay requirements all information bursts will be short – thus short signalling time is vital. The minimum signalling time in a transmission is the sum of the preamble, SOM and Par, adding up to ≈ 5.2 ms in a system utilising the lowest data rate. A self-contained signalling message must contain some additional information. Using a 4-5 ms long interleaver, the shortest possible signalling transmission lasts approximately 10 ms. Using the lowest rate (30 ksymbols/s) and a ⅓ rate coder this gives 40-50 bits to the link layer.

An efficient link protocol will use cross-layer design and adapt to the underlying physical layer constraints as well as the functional requirements.

3.0   DESIGN PRIORITIES AND TRADE-OFFS

Since the waveform (at least in the first level of ambition) is restricted to 25 kHz bandwidth, the channel capacity will be limited. Traditional CNRs have offered a maximum data rate of 16 kbps. Modern radios offer data rates up to approximately 50-100 kbps, and so should NBWF. But, we must be aware that the higher data rates require a larger signal to noise ratio and thus will have a sufficiently reduced range. It is thus anticipated that in many situations the lowest data rate will be used, giving a transmission range comparable to traditional CNRs.

Much published work in recent years has focused on multi-hop networks. One might argue that instead of using low rate single hop it would be more efficient to use higher data rates in a multi-hop network. This might be true for unicast transmission under ideal conditions, but a dynamical (mobile) multi-hop network requires more signalling and leads to more erroneous routing than a single-hop network which is more static at the same level of mobility. It is shown in [7] that higher data rates (and thereby reduced transmission range) do not necessarily provide better performance in a multi-hop mobile network.

Another aspect is that much of the traffic is radio-broadcast or multicast of nature. For such traffic it may be shown that for most topologies, a single long-range transmission at a low data rate is more efficient than the multiple transmissions required at a higher data rate (and shorter range) in order to reach the same set of destinations.

For these reasons we have optimised our link protocol based on using the lowest data rate of 20 kbps, since experience shows that a long range is usually required. In situations with geographically dense networks it may be possible to use a higher data rate and still maintain a more or less single hop network. For unicast traffic a rate adaptation algorithm should be used to obtain a higher throughput.

The significance of the multicast PTT voice application with its strong delay and jitter requirements simplifies the choice of class for the MAC protocol. These requirements will be difficult, if not impossible, to fulfil with any kind of random access protocol. For that reason we have chosen to use a type of TDMA protocol, enabling resource reservation for multicast voice.

Although multicast voice is important, we see a growing number of applications requiring data transfer. As the required data capacity is highly variable and not very predictable for many applications, we need a MAC protocol that can support this. Due to the growing capacity demand compared to the available data rate, we have tried to minimise the required permanent resource allocation for voice.

So, we have ended up designing a dynamic TDMA protocol that can support all types of services: voice, predictable data services, and services with a more unpredictable capacity requirement. We have focused our design on making the protocol very dynamic, with short and efficient signalling for reservation of resources.

4.0   OUTLINE OF THE MAC PROTOCOL

Our task is to provide an efficient solution for MAC to integrate voice, data and RBCI within 25 kHz bandwidth. This should work also at the lowest (and primary) data rate of 20 kbps. For multicast voice we prefer ≈ 200 ms transmit buffering as a compromise between the delay requirement and the protocol efficiency. This also becomes the TDMA frame length (or cycle time). The voice application could benefit from shorter buffering, but that would result in short and inefficient transmission lengths for data.

We have chosen the TDMA frame length (202.5 ms) to be an integral multiple of the MELPe frame length since the voice application is the most important one for this kind of networks. A selection of 9 slots in a TDMA frame should give a sufficient flexibility in transmission length. In order to be able to allocate a small basic transmission capacity for each radio we propose a superframe structure on top of this. The number of frames in a superframe may be flexible, depending on the actual number of radios in the network and the required "static" capacity for each radio. Two or more slots in each frame are suggested to be allocated on a fixed basis to be shared between the radios through this superframe structure. The TDMA structure is shown in figure 2.
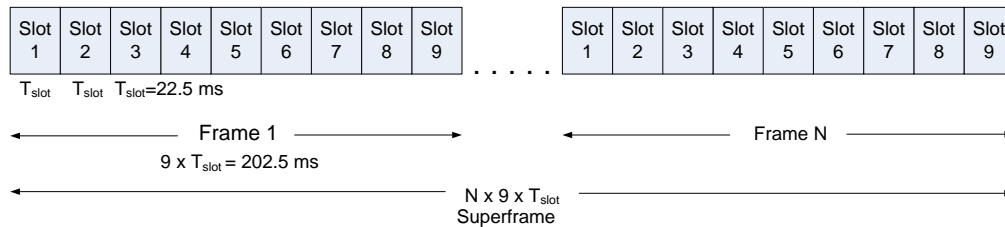

Figure 2:   TDMA slot and frame structure

If we consider transmission in a single slot at a PHY rate of 20 kbps and subtract all PHY overhead[3], we find that each data transmission lasts about 14 ms and contains a total of 280 bits for the link layer including PCI. If we allocate 150 bits of link PCI per transmission, this leaves us with only 130 bits of user data (including any higher layer PCI) in each transmission. A single slot per frame will give a data rate of only 640 bps. For this reason, we propose slot merging to be used whenever possible. Two merged slots per frame results in a link payload throughput of 2.8 kbps. Figure 3 shows the link layer throughput as a function of number of merged slots per frame, for all available channel rates.

A voice connection using MELPe at 2.4 kbps requires the allocation of 2 slots per frame for each radio hop at a PHY rate of 20 kbps. A multicast voice transmission that requires an additional relay ties up a total of 4 slots per frame. As MAC should be able to support more than one simultaneous voice transmission, e.g. due to relaying, this would normally require the reservation of many slots on a permanent basis in order to be able to support Push-to-talk without delay. In a network with a basic data rate of 20 kbps this would in practice leave only half or less for data applications.

For this reason we have pursued a solution where channel resources for voice are primarily reserved on demand. This is done through a procedure that is designed to be rapid but not 100% fail proof. What

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[3] This also includes time for synchronisation, propagation, rx/tx switching, amplifier ramp-up/ramp-down and receiver AGC.

makes it applicable is the predominantly single-hop nature of CNR networks, at least at the lowest transmission rate.
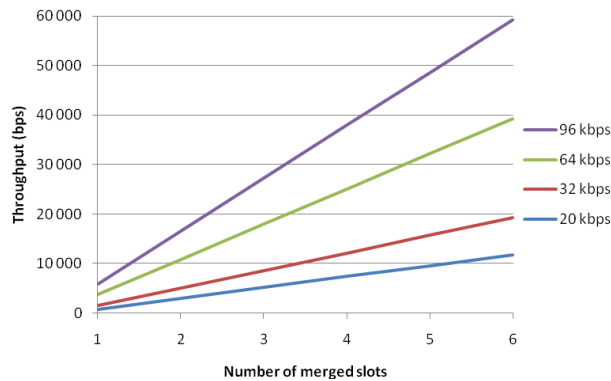

Figure 3: Link throughput as a function of number of merged slots.

Channel resources are reserved by the radios upon demand through a distributed algorithm. Reservation for multicast voice receives special treatment for two reasons: 1) due to the connection setup delay requirement and 2) due to the nature of PTT voice which generally implies only one user speaking at a given time.

Some basic principles apply for slot usage:

- Unreserved slots are open for contention, regulated by type of service, priority and traffic load

- Multicast voice has priority over data (within certain limits). Reserved slots can not be accessed by other radios. Pre-emption must be signalled in other time slots.[4]

Due to the multicast voice (MV) delay requirement we will have to reserve one slot on a permanent basis, in order to guarantee the necessary capacity for the signalling required to set up an MV channel. This slot is used to transmit voice whenever voice is active. One additional slot should be reserved for each relay. By not allocating the maximum required capacity on a permanent basis, we may not guarantee the setup delay or the successful set up of a PTT voice call when requested. But the alternative leads to a significantly reduced data capacity, especially when relaying is allowed.

5.0 OUTLINE OF THE LLC PROTOCOL

The main tasks for the Logical Link Control (LLC) layer are: resource reservation; segmentation of long packets onto a number of self-contained transmissions; and the ARQ protocol to ensure reliable link transport. In addition, LLC performs relaying of multicast voice. (Relaying of data is performed at the network level.)

5.1 Reservation and contention

MAC offers two basic transport services for data: access to reserved time slots and contention access to unreserved time slots. Through the superframe structure, each radio is allocated access to a few time slots, repeated with an interval of some frame lengths (depending on the number of radios sharing the channel). But this fixed capacity is not well suited for low latency services and dynamic capacity demand.

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[4] In order to achieve a high utilisation of the PHY rate with short burst transmissions.

We propose a distributed resource reservation mechanism with efficient signalling for multicast voice. The initiator will transmit a connect request (CR) message, requesting its reservation to be confirmed by one or more selected neighbour(s). The selected neighbour(s) will respond by each transmitting a very short connect confirm (CC) message indicating whether resources are available or not. The purpose is to inform hidden nodes in multi-hop networks. This requires a lot of signalling to set up an MV connection. But we do not require reception of positive CCs in order to complete the MV setup, as packet loss then could reduce the success rate.

The same mechanism is used to offer a reliable data service based on reservation. A radio is allowed to allocate all unreserved time slots for a limited time, typically the time required to transmit an IP packet. The reservation process must use already reserved (superframe) slots or contend with other radios in unreserved slots.

When transmitting data, LLC must determine which type of data service to use, depending on delay requirement and packet size: contention, superframe allocation or reservation. Signalling for reservation is shown in figure 5.

5.2 Segmentation

An IP packet received over the Ethernet connection typically has a maximum size of 1500 bytes. Even when transmitting at 96 kbps, 6 consecutive time slots are required to complete the radio transmission. This means that we need a segmentation function for all the other cases (lower rate or fewer slots available). A data transmission using 2 slots at 20 kbps contains a link payload of about 75 bytes. This means that a maximum length IP packet in such cases must be transmitted as more than 20 segments, which results in a total duration of more than 4 seconds.

There are two factors that influence the segment size: the transmission rate and the number of slots available for the data transmission. Both factors may change during the transmission of an IP packet. The rate may be reduced due to poor transmission conditions and the number of slots due to pre-emption by multicast voice (see figure 4). On this background we propose to use a fully flexible segmentation that can utilise any available combination of transmission time and rate. The resolution of the segment size will be 1 byte in order to achieve full flexibility (with some added cost for PCI).
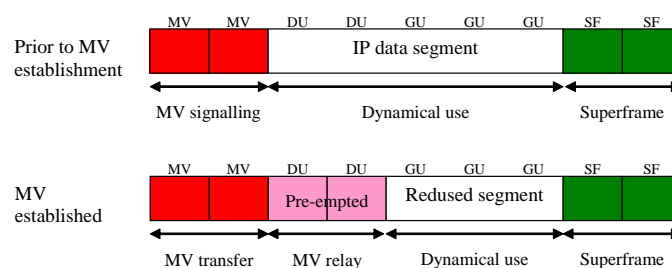

Figure 4: Dynamic segmentation is required due to MV pre-emption.

5.3 Selective repeat ARQ

We propose to use a selective repeat ARQ mechanism for the segments of an IP packet, as shown in figure 5. The source, which has reserved the time slots, determines when it expects an acknowledgement. It will then reserve some of its own capacity to be used by the destination for acknowledgement. In addition to handling packets consisting of a number of segments, the ARQ protocol must be able to handle the fact that when re-transmitting a segment, the transmission rate or the available transmission duration may have

changed. One retransmission may actually consist of e.g. two shorter transmissions if the data reservation has been pre-empted by MV. This could also happen if the transmission conditions turn out to be poorer than estimated, resulting in a reduction in transmission rate at the time of re-transmission. The number of segments transmitted before acknowledgement is determined by the source, based on its expected channel quality.
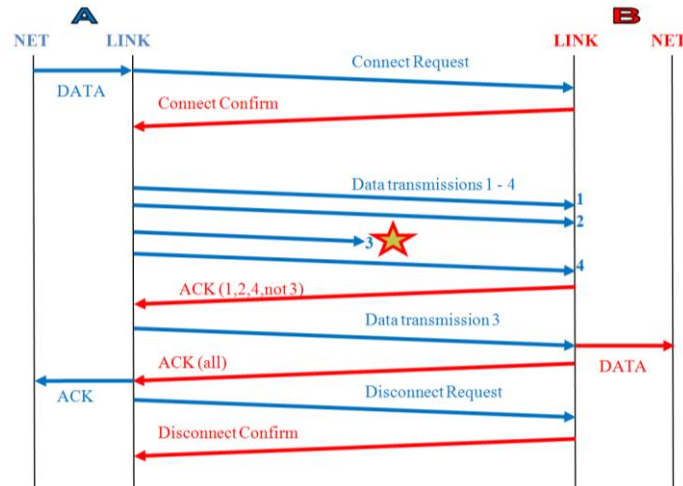

Figure 5: IP data; reservation signalling and the selective repeat ARQ protocol.

6.0   PROTOCOL PERFORMANCE

In this section, two simple simulation studies are presented. Simulations are carried out using an NBWF model based on the Omnet++ [8] network simulator. The model includes a simple representation of the proposed physical layer, as well as the link layer described in the previous sections. Refer to [9] and [10] for a detailed description of the simulator.

6.1   Multicast Voice

Since we have chosen a dynamic reservation mechanism for multicast voice, we have to pay a delay penalty. Figure 6 shows the signalling in different time slots and the associated voice delay, which is 200-400 ms longer than a system with fixed reservation. The setup delay is confirmed by simulations.

In order to achieve protection against collisions with data transmissions, we propose to send MV CR and CC messages only in the pre-allocated signalling slots as described in figure 6. An alternative scheme would be to allow scheduling of CCs in the yellow and green slots if not in use, in order to reduce the connection setup delay. However, due to the dynamic and unpredictable nature of multi-hop radio networks, the nodes may have different view on the current slot usage. Specifically, the presence of hidden nodes[5] will dramatically reduce the connection success probability if using the alternative method as shown in the following simulation study.

Figure 7 shows the network topology used for our simulation study. The connection between two adjacent nodes denotes that they are within each other's demodulation range (i.e. one-hop neighbours). For nodes within the demodulation range, the probability of successful reception of a transmission is a function of

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[5] A node is hidden with respect to a given connection, if it is unable to receive any of the CRs for that connection, but is able to disturb any of the nodes that can receive the CR.

the SNR, which degradation is caused by path-loss (using the Egli [11] propagation model) and collisions. Thus, all collisions are not destructive.

Figure 6:   Case with relay; signalling for multicast voice and analytical estimation of setup delay.
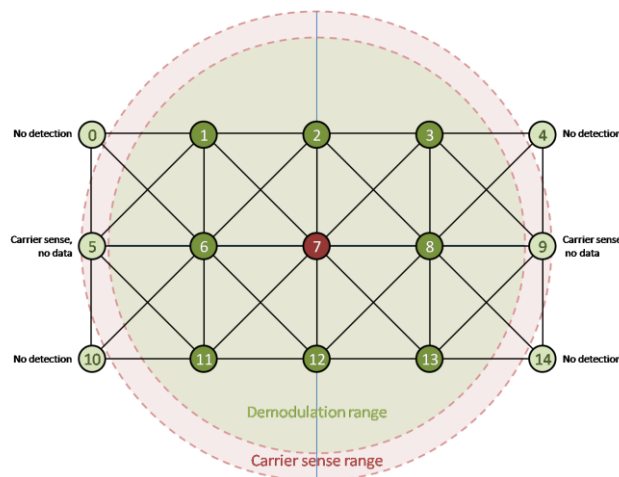

Figure 7:   Network topology. Red node: MV relay. Dark green nodes: MV group members. Light green nodes: "hidden nodes". The dashed circles illustrate the relay's demodulation and carrier sense ranges.

A five second MV call is initiated deterministically each 8th second from one of the group members selected uniformly. All nodes generate background data traffic with negative exponentially distributed (n.e.d.) arrival distribution, and packet size of 1480 bytes (eight segments when 4 slots are available). The background traffic is transmitted only in the dual use (DU) and general use (GU) slots, using the contention based reservation access method for CR as described in section 6.2. Both the proposed and the alternative schemes are simulated and the focus is on the performance of node 3 which is degraded by the hidden nodes 4 and 9.

Figure 8 clearly illustrates the advantage of only sending the CC messages in the pre-allocated signalling slots. When using the proposed scheme, the connection setup at node 3 is successful for all calls, while a significant performance reduction is seen when using the alternative method as the background traffic load increases.

As expected, this gain in robustness comes with a MV delay cost. Figure 9 confirms that the delay difference between the two methods is approximately 200 ms. When using the proposed scheme, the voice transmission does not start until frame 3, while the alternative scheme allows voice transmissions in frame

2 (see figure 6). Note that the voice delay is independent of the traffic load, which is an important feature of the protocol. This is because ongoing data connections are pre-empted when a MV call is initiated. As a consequence, the delay for the background data traffic increases significantly with the load.
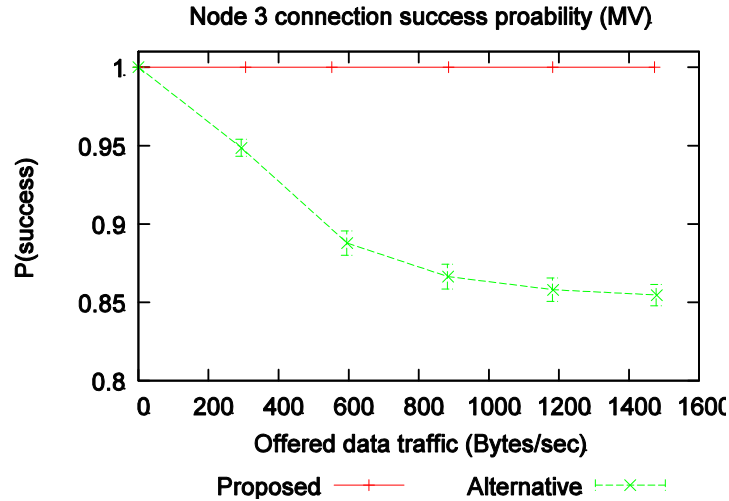
Node 3 connection success proability (MV)

Figure 8: MV connection success probability, defined as the probability that a group member receives the CR and one of the first two voice PDU.
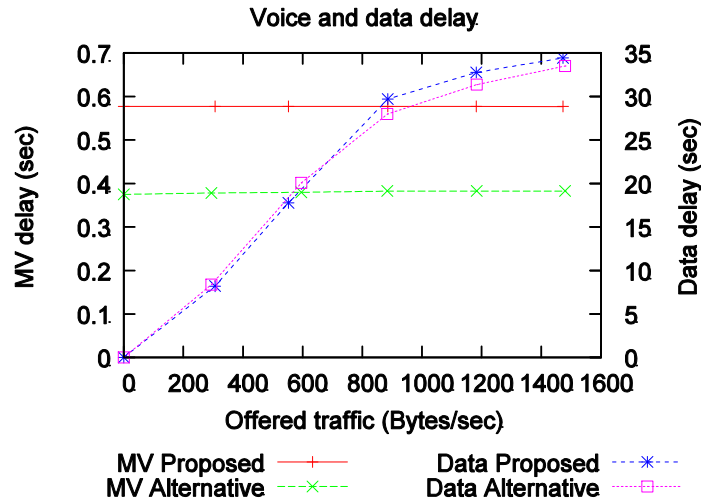
Voice and data delay

Figure 9: Mean MV delay (left axis): The time duration from the PTT event to the arrival of the first voice PDU at the receiver. Data delay (right axis): The mean layer 7 one-way delay.

6.2 Reservation access for data

In order to reserve time slots for the transfer of a data packet, a reservation protocol similar to MV is used. For data however, the signalling messages are sent in unreserved slots, thus requiring the use of a contention access scheme. In the following, we study a simple contention access method where the contention window consists of all DU and GU slots (see figure 4).

At the start of the window, each node having a data packet for transmission draws a random delay from a continuous uniform distribution, ignoring the time slot boundaries. If no preamble is detected before this

time, the node sends a CR message. The destination node (assuming unicast) responds with a negative or positive CC. If positive, the originator node sends one segment in each TDMA frame until all segments are transmitted.

Both a multi-hop and a single-hop topology are studied. The multi-hop topology is identical to the one used in the previous simulation study. The data traffic generator is also the same, while the MV generator is omitted. In the single-hop topology, all the 15 nodes are within each other's demodulation (and carrier sense) range. A fundamental difference between the single and multi hop scenarios is the collision sensitivity period, which in the single-hop case only includes the length of the preamble. In the multi-hop case however, the sensitivity period starts when the CR is sent and ends when the destination replies with the CC. If the responder receives a preamble from a hidden node during this interval, the connection will fail.
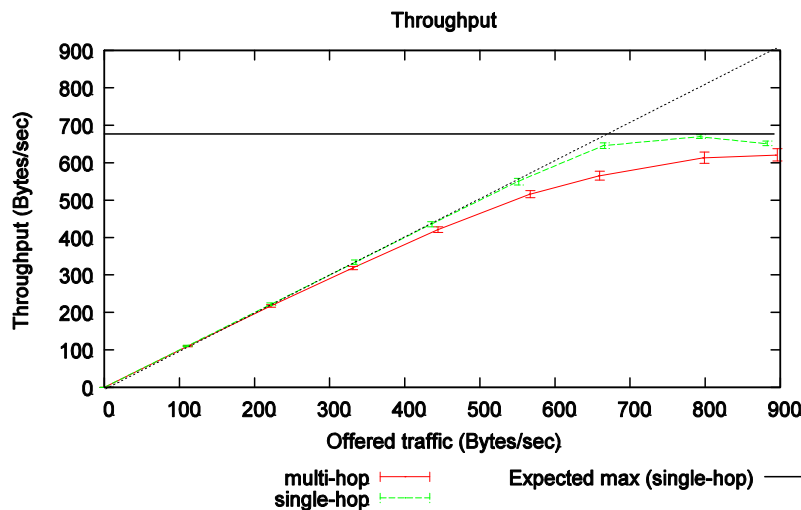

Figure 10: Random access data performance.

Figure 10 shows the throughput as a function of offered traffic for both topologies. The horizontal line marks the expected theoretical throughput in the single-hop case when all nodes always have at least one queued packet. We observe that the access method performs well in the single-hop case, as the throughput follows the upper bound closely. In the multi-hop case, the protocol also performs near optimally when offered load is below 400B/s. Above this point; the increased sensitivity period due to hidden nodes unavoidably causes some reduction in the performance relative to the single-hop case. Further work will address this issue, presumably by adapting the contention window to the traffic load.

7.0 CONCLUSION

We are proposing a link layer (MAC and LLC) to be used in a new NATO Narrowband Waveform standard. Within 25 kHz the PHY supports data rates from 20 to 96 kbps. Based on a dynamic TDMA protocol, the link layer shall support concurrent IP data and multiple multicast voice connections, in addition to Radio Based Combat Identification (RBCI). Dynamic resource reservation is supported for both IP data and multicast voice. Our ambition is to support automatic relaying not only for IP data, but also for multicast voice.

Our link layer proposal is designed to be dynamic and flexible. Compared to existing Voice+Data protocols that rely on a less dynamic reservation scheme, we achieve a higher average IP data throughput

in a typical multi-hop scenario with fluctuating traffic load. The penalty is some extra delay for multicast voice, which in contrast to comparable systems relies on reservation and pre-emption.

Simulation results for multicast voice confirm the added voice delay, and indicates that pre-emption of hidden nodes should receive extra attention in order to find the best solution. Some initial performance results for the IP data protocol are also available. The reservation protocol for IP data seems to perform reasonably well according to our theoretical expectations. In addition, it shows only a minor degradation in a multi-hop network compared to a single-hop network.

Future simulations will compare our reservation mechanism to a traditional contention mechanism for IP data transmission. The reservation mechanism should perform better than random access in a multihop environment. In a single-hop network we do not necessarily expect that to be true.

Our work is not yet complete. Even though the first results are promising, there are a number of details and parameters that need tuning for performance optimisation. The primary objective is to achieve an acceptable performance for multicast voice with regard to setup and voice quality. Reliable delivery and throughput for IP data is our second objective. One important aspect for further study is the MV pre-emption of IP connections. Successful pre-emption relies on the IP reserving node to be informed either by the MV reserving node directly or through one of the CC nodes. This may not work perfectly in a multi-hop network with packet loss due to external noise. We have to study this through simulations to determine how often it fails and find the best way of selecting the CC nodes for MV.

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